# EXHIBIT 1

| Uppgjord – Prepared | | Datum – Date | Rev | Dok nr – Doc no |
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| Paul W. Dent | 9199907121 | 1994-10-27 | | T/V 94:0020 |
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## INVENTION DISCLOSURE FOR:

## DUAL MODE SATELLITE/CELLULAR TERMINAL

### 1. FIELD

The invention pertains to mobile or portable wireless telephones that can operate either through land-based cellular systems or through orbiting satellites, if no land-cellular base station is within range, and in particular to means for choosing satellite signal formats to facilitate re-use of components in wireless telephones when switching between modes so as to reduce cost.

### 2. SUMMARY

This application is a Continuation-in-Part of U.S. patent application no. .......... entitled "Satellite/cellular telephone" (Dent, filed .........) (BDSM CASE 027545-020), which is the parent application to this application and thus forms part of this application in its entirety.

A portable wireless terminal is now disclosed having means to operate according to a known digital cellular standard such as GSM, such means comprising receiver radio frequency components for receiving a TDMA burst and digitizing it and signal processing components for decoding it and reconstituting a voice or data signal. The inventive terminal uses the same receiver components to receive a satellite TDMA burst that preferably employs the same bitrate and format but occurs less frequently due to the digital voice signal from the satellite being encoded at a lower bitrate.
The terminal transmits a TDMA burst at a submultiple of the receive bitrate for a proportionally longer time using a transmit frequency channel that is proportionally narrower in bandwidth. The transmit timeslot and transmit frequency channel allocation are linked to the receive frequency and timeslot allocation in such a way that transmission and reception at the terminal do not overlap and have an almost constant relative timing relationship as determined by a timing controller that compensates for loop propagation delay.

### 3. PRIOR ART

The signal bandwidth and channel spacing used for satellite communication is generally different from the signal bandwidth used in cellular systems. One reason for this difference is that satellite communication is thermal noise limited, favoring lower bandwidth and coding rates, while cellular is interference limited, favoring higher bandwidth and coding rates.

For example, the GSM cellular system's channel spacing is 200KHz, while the INMARSAT-M satellite system uses 5KHz or 10KHz channel spacing. In the latter narrowband mode, frequency and phase noise is considerably more troublesome

for moving terminals than in the former, wideband mode. There can thus arise difficulties in attempting for economy in the terminal to re-use circuitry for both modes.

The parent application describes a dual-mode terminal equipped with a novel frequency synthesizer circuit that economically provides both narrowband satellite channel spacings as well a GSM spacings while meeting the stringent noise requirements in a narrowband satellite mode and the fast switching requirements for the GSM frequency-hopping mode.

Other prior art descriptions of dual mode terminals that economically re-use the same components between two modes may be found, for example, U.S. Patent Application 07/585,910 entitled " Multi-mode signal processing" (Ekelund and Dent, filed .........) describes re-using the same components to process alternatively an analog FM signal according to the AMPS cellular standard or a digital cellular signal according to TIA standard IS54.
The above application is incorporated by reference herein along with the published standard for the Global System for Mobile communications known as GSM.

The GSM standard itself discloses the possibility to transmit lower bitrates by transmitting TDMA bursts at the same bitrate only less often. The GSM standard describes a so-called "half-rate" mode in which a burst is transmitted only every 16 timeslots instead of every eight. However, the same format is used in the uplink direction (mobile to base) as in the downlink direction (base to mobile), which leads to problems of high peak power requirements from the mobile phone in a satellite system.

U.S. patent application no. 08/179,954 (Dent, filed 11 Jan 1994) discloses assymetrical TDMA formats in which uplink TDMA formats can have a smaller number of timeslots combined with a greater availability of narrower bandwidth frequency channels thna the corresponding downlink TDMA formats, thus reducing the peak-to-mean power ratio needed in the mobile terminal. This patent application is also incorporated by reference herein. When practicing the invention disclosed in the above incorporated application however, a terminal is not capable of being compatible with the GSM cellular standard's uplink waveform.

The above deficiencies of the prior art are surmounted and other improvements to the prior art are obtained when practising the invention further elaborated and claimed herein.

## 4. DESCRIPTION OF THE INVENTION

Figure 1 shows the allocations of frequencies for Personal Satellite Communications services in world regions R1,R2 and R3 compared to FCC proposals for new frequencies to be offered by auction for new landbased Personal Communications Services. It can be seen that the PCS bands marked DEFG conflict with the sattelite PSC bands; however, the FCC has at the

present time abandoned plans to allocate frequency bands DEFG to PCS, and the planned frequency auction is restricted to bands marked A,B and C. The separate A,B and C groups represent up- and downlink bands for directions mobile to base and base to mobile respectively, and the 80MHz separation is known as the duplex spacing.

In between the up- and downlink bands, the frequencies will be offered on an unlicensed and largely unregulated basis. The unregulated band has no envisaged frequency duplex spacing and is only suitable for simplex, half-duplex or press-to-talk systems, or systems which use same-frequency time-duplex operation such as the Digital European Cordless Telephone Standard (DECT).

The duplex spacing between satellite up- and downlink bands is seen to be somewhat larger. While this facilitates the construction of small, low-loss duplexers that would permit transmission and reception through the same antennas simultaneously, duplexers are still components that are preferably avoided by using the time-duplex method at least in the portable telephone terminal. Avoiding such highly frequency-selective components also facilitates the construction of a receiver that would embrace both the PCS receive band 1930-1970 MHz as well as the PSC receive band .

The invention includes, but is not restricted to, the construction of dual-mode PCS/PSC terminals in the bands illustrated in figure 1. The invention can alternatively be applied to dual-mode terminals in which the cellular band is in the 900MHz range, or indeed in which the satellite and cellular frequency bands lie in any frequency range, either the same or different.

A satellite system at the present state of the art cannot approach the capacity of a landbased cellular system to serve millions of subscribers. An objective of providing dual-mode cellular satellite terminals is thus to ensure that subscribers use the high capacity cellular system wherever it is available, such that only subscribers that are temporarily outside of cellular coverage have to employ the limited capacity satellite.

A satellite system can provide global coverage however, and thus the principal reason for transferring a call to the satellite system occurs when the subscriber has travelled to a country which has a non-compatible cellular system.

The satellite loading may indeed be dominated by that category of subscribers dubbed "the travelling businessman" who is temporarily outside the native cellular system for which his phone is designed to operate, albeit being within the cellular coverage of a non-compatible foreign system. Such subscribers can still receive service using the satellite mode of the inventive dual-mode phone described herein.

In Europe, the PCS frequency bands that have been allocated are slightly lower in frequency than the U.S. PCS bands, and the duplex spacing is 95MHz as opposed to 80MHz. The European PCS system is known as DCS1800 and uses the 900MHz GSM standard translated to the higher frequency bands.

Avoidance of very frequency selective duplexing filters also facilitates the construction of a terminal that operates in both U.S. and European PCS bands. Thus it is even possible to conceive of a PCS/PSC terminal that is at home in both the U.S. PCS and European DCS1800 systems, thus avoiding loading the limited satellite capacity still further. For administrative reasons, such as billing the subscriber for services, such a phone may be designed to store up to three sets of subscription data, for example a subscription with an operator of the U.S. PCS system, a subscription with a European DCS1800 or GSM system operator, and a subscription in the global satellite system. The GSM standard describes the facility to store subscription information including security and authentication keys external to the phone in a plug-in "smart card". One embodiment of the invention can comprise using a smart card that not only contains alternative subscription information that is electronically read into the phone, but also a description of the alternative mode and therefore signal waveforms which the phone's signal processing will adopt when that subscription data is in use.

For subscriber and retailing convenience however, the preferred embodiment is to adopt the same set of subscription data including telephone number and to ensure that that data will be accepted as valid in all systems when the subscriber is roaming between systems.

Avoiding loading the satellite system unnecessarily can actually enhance the satellite operator's revenue, as subscriptions can be sold and fees collected from a greater number of subscribers without risking system saturation. Subscription revenue can greatly exceed call charges when the total number of subscribers is much greater than the number instantaneously making calls through the satellite system. Thus it is not necessary to charge a premium to subscribers that are temporarily outside cellular coverage and who must be connected via a satellite. Call charges and billing can thus remain at the same levels irrespective of how the service is delivered, i.e. via a satellite or via a land network, so that the use of the satellite or landbased network is completely transparent to the subscriber.

Figure 2 illustrates the TDMA transmission format employed in GSM at both 900MHz and 1800MHz bands.

A superframe comprises 4x26 TDMA frames. In every 26 successive TDMA frames, the first 12 carry traffic information; frame 13 is idle and can be used by a mobile terminal to read identification data from nearby base stations; frames 14-25 carry traffic information, and frame 26 carries one fourth of a Slow Associated Control Channel message (SACCH). Four such blocks of 26 frames are required to complete delivery of one SACCH message, while each group of 26 frames delivers traffic data representing six, 20mS speech vocoder data blocks. Each 20mS block of coded data representing a segment of the speech waveform is spread over eight consecutive TDMA frames in a process known as block-diagonal interleaving. Each 8-frame interleaved block is half-overlapped and merged with 4 frames of each of the adjacent speech blocks in order to fill each timeslot with bits that have come half

from one speech frame and half from another. Each TDMA frame is then transmitted on a different frequency using frequency hopping to obtain the benefit known as interference averaging or interferer diversity.

Each TDMA frame lasting approximately 4.615mS is divided into 8 timeslots. One mobile signal utilizes only one of the eight timeslots in each frame, and other mobiles use the others. Figure 1 also shows the internal structure of data symbols within each slot. A 26-bit syncword of known symbols lies in the center of the slot and is used to determine the characteristics of the transmission channel and to train an equalizer to perform optimum demodulation. On either side of the syncword, two flag bits are placed. The flag bits from eight consecutive frames over which a speech block is interleaved are majority combined to form an indication of whether the interleaved block is speech, or a Fast Associated Control Channel (FACCH) message. A speech block may be stolen to send an urgent FACCH message, indicating for example that the mobile has reached edge of cell and is to be handed off to an adjacent cell, and the flag bits indicate to the phone when a block has been stolen for FACCH.

On either side of the Flag bits lie 57-bits of data that may form part of a speech block or FACCH message as described above. Half of the 114 bits belong to one speech or FACCH block and the other half to an adjacent interleaved block. At each end of the slot, tail bits are added. The tail bit periods are used partly to terminate the equalizing demodulator operation cleanly when equalizing for delayed echos, and partly to permit the transmitter power to be smoothly ramped up and down to avoid spectral spreading into adjacent frequency channels.

One coded speech frame thus consists of half of the 114 bits from 8 consecutive slots, that is 456 bits per 20mS or 22.8 kilobits per second on average. A portion of the 456 bits represents perceptually important speech bits transmitted with a 2:1 redundancy for error protection. A rate 1/2 convolutional code is used for this purpose. After decoding, the 2:1 redundancy is removed for this portion of the bits and the decoded bitrate is then an average of 13kilobits/second. The speech vocoder operates at the 13KB/s rate, which provides good sound quality, even with non-speech input sounds such a background noise.

A satellite system normally requires to sacrifice some of the advantages of high quality vocoders, namely the just-mentioned robustness against non-speech background noise, in order to reduce the information rate transmitted and thus conserve satellite and mobile battery power. Power is more critical for communicating with a satellite because of the great distances involved. Typically, a satellite communications system may employ a speech vocoder operating at 4Kilobits per second. When noise limited rather than interference limited, it is not advantageous to apply error correction coding only to a portion of the speech bits, so a rate 1/2 code may be applied to the whole of the 4kilobits per second, raising it to a coded bitrate of 8 kB/S. This is around 1/3rd of the GSM coded information rate.
The lower bitrate could be transmitted from the satellite to the mobile using a proportionally lower bandwidth. This is undesirable however as the

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filter components used to establish receiver bandwidths are large and costly, and it is one objective of this invention to avoid duplication of receiver filters for a cellular bandwidth and a different satellite bandwidth.

Alternatively, the reduced satellite bitrate could be transmitted using the same bandwidth but in a shorter timeslot containing fewer bits. The number of timeslots in the TDMA frame could then be increased to serve other mobile conversations. The energy used per conversation thus reduces, which is the objective of using the lower rate vocoder. A shorter timeslot containing fewer information bits is however not desirable, as the overhead of the sync word, flag bits and tail bits would be a higher proportion of the total throughput, leading to inefficiency and loss of capacity. Therefore the invention comprises instead transmitting the same number of bits per slot but increasing the time between slots to reduce the average bitrate, i.e. to increase the TDMA frame period by increasing the number of slots.

The factor by which the number of slots in the frame period is increased must clearly be a small integer. The effect of this factor on raw bitrate and the number of signals accommodated per 200KHz carrier is shown in the table below:

| FACTOR | FRAME LENGTH (SLOTS) | RAW BITRATE (KB/S) | |
|---|---|---|---|
| 1 | 8 | 22.8 | (GSM) |
| 2 | 16 | 11.4 | (GSM "HALFRATE") |
| 3 | 24 | 7.6 | |
| 4 | 32 | 5.7 | |

As the number of slots in the frame increases, it may appear that the capacity of the system is increased; however, this ignores the effect of cochannel interference. When less coding is provided, the tolerance to interference is lower and it is necessary to increase the distance between mobiles using the same channel, thus allowing frequency re-use only on a sparser grid.
This trade-off is more specifically addressed in U.S. patent application no. ......... entitled "Communications System using Cochannel Interference Cancellation" (Dent, filed Jan 11 1994) which is hereby incorporated by reference in its entirety. The trade-off is repeated here in order to illustrate how the frame length (in slots) is chosen.

Clark and Cain "Error Correction coding for Digital Communications gives the required Eb/Nos for 0.1% BER for constraint length 6 convolutional code rates of 1,3/4,2/3,1/2 and 1/3 as follows:

r Eb/No for BER=0.1%

| | |
|---|---|
| 1 | 6.7dB |
| 3/4 | 3.9dB |
| 2/3 | 3.5dB |
| 1/2 | 3.0dB |
| 1/3 | 2.6dB |

Values for lower rates are of 1/4 and 1/5th are estimated by extrapolation. These figures are without interference, and must be increased if cochannel interference is present at levels described by the Carrier to Interference ratio (C/I). The C/I that requires an increase of Eb/No of respectively 0.5, 1 and 3dB to compensate for the interference is given in the table below:-

| REQUIRED C/I for | 0.5dB loss | | 1.0dB loss | | 3.0dB loss | |
|---|---|---|---|---|---|---|
| | BPSK | QPSK | BPSK | QPSK | BPSK | QPSK |
| Coding rate 1 (none) | 17.2dB | 20.2dB | 13.7 | 16.7 | 9.7 | 12.7 |
| 3/4 | 13.2 | 16.2 | 10.9 | 13.9 | 6.9 | 9.9 |
| 2/3 | 12.2 | 15.2 | 8.7 | 11.7 | 4.7 | 7.7 |
| 1/2 | 10.5 | 13.5 | 7.0 | 10.0 | 3.0 | 6.0 |
| 1/3 | 8.3 | 11.3 | 4.8 | 7.8 | 0.8 | 3.8 |
| 1/4 | 6.8 | 9.8 | 3.3 | 6.3 | -0.7 | 2.3 |
| 1/5 | 5.7 | 8.7 | 2.2 | 5.2 | -1.8 | 1.2dB |

etc

It may be seen that while the Eb/No needed for a given error rate planes out with increasing coding, the C/I requirement becomes continuously more relaxed as the coding is increased due to the steadily increasing bandwidth, which provides an increasing spread-spectrum processing gain upon decoding.

The above results for the static channel are pessimistic for fading channels. When Rician or Rayleigh fading is present, the mean Eb/No must be increased above the static Eb/No requirement to maintain the same error rate. However, on the satellite downlink, the C/I ratio does not exhibit fading, because both the I and C reach a given mobile over exactly the same channel and fade by exactly equal amounts. Thus the C/I does not reduce 10dB when the Eb/No fades 10dB, but stays at the original value. Since most of the errors occur when the Eb/No fades much below its mean value, the effect of additive C/I of 10dB at that point is not so important. This has been verified by simulations performed during reduction to practice.

Returning to the choice of frame length factor equal to 1,2,3 or 4, these numbers equate to coding rates of approximately 4/22.8 = 1/5 approx; 4/11.4 = 1/3 approx; 4/7.6 = 1/2 approx, and 4/5.7 = rate 2/3 approx, given a 4kilobit uncoded voice bitrate to be transmitted.

| Uppgjord - Prepared | | Datum - Date | Rev | Dok nr - Doc no |
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From the above table therefore, we can see that the C/I needed for less than 0.5dB degradation of the Eb/No power budget, using QPSK modulation in a static Gaussian noise channel would be approximately as shown in the table below:

| Frame length in slots | C/I for QPSK and < 0.5dB Eb/No loss | C/I for IM = -20dB | C/I for -20dB IM and -16dB adj. ch. |
|---|---|---|---|
| 8 | 8.7 | 9.1 | 10.01 |
| 16 | 11.3 | 12.0 | 14.09 |
| 24 | 13.5 | 14.6 | 20.2 |
| 32 | 15.2 | 16.6 | not satisfied |

| Frame length in slots | C/I for QPSK and < 1.0dB Eb/No loss | C/I for IM = -20dB | C/I for -20dB IM and -16dB adj. ch. |
|---|---|---|---|
| 8 | 5.2 | 5.35 | 5.7 |
| 16 | 7.8 | 8.07 | 8.8 |
| 24 | 10.0 | 10.46 | 11.9 |
| 32 | 11.7 | 12.4 | 14.9 |

It is thus seen that increasing the number of slots, while appearing to increase the capacity, also increases the C/I requirement, which means that a greater distance must be maintained between co-channel users, thus decreasing the area density of conversations per Megahertz.

Another source of co-channel interference is intermodulation in the satellite transmitter power amplifiers. IM may be reduced, but only at the expense of power conversion efficiency from expensive solar-cell generated DC power to radio frequency communications power. Using IM reduction techniques disclosed in U.S. patent application no........... entitled "Improved Matrix Power Amplifier??" (Dent and Lampe, filed Jan 11, 1994) which is hereby incorporated by reference, it is possible to obtain -20dB IM at zero dB of input backoff, by which is meant that the transmitter power amplifiers saturate already at an instantaneous signal level equal to the rms value.

The C/I needed when IM of -20dB is present is indicated in the second last column of the above table.

Another source of interference is adjacent channel interference. The GSM modulation is Gaussian Minimum Shift Keying (GMSK) with a Gaussian filter having a BT product of 0.3. This is described more fully in the GSM specifications. GMSK(BT=0.3) modulation leads to energy in the adjacent channels +/- 200KHz away that is around 18-20dB down on the main lobe energy. As long as the satellite radiates adjacent channel signals at the same power level as a wanted signal in between, the sum of the interference from both adjacent channels would be between -15 and -17dB relative to the wanted signal. Using a median value of -16dB, the last column in the above table gives the co-channel C/I needed with both PA intermodulation of -20dB and adjacent channel interference totally -16dB. This indicates that the 32-timeslot option no longer meets the required signal quality with only an

0.5dB loss of Eb/No performance, due to insufficient coding. The second table shows the C/I needed for a 1dB loss of Eb/No performance, showing that 32 slots can still in principle be accommodated.

In practice, there are two factors mitigating the effect of interference:

(1) The already mentioned fact that co-channel C/I does not vary with fading, and

(2) The use of Discontinuous transmission, which means that half of the co-channel and adjacent channel interferers are momentarily silent.

Against the above mitigating factors however, the assumption that adjacent channel interferers are at the same level as the wanted signal may not be true. It is desirable in a satellite or cellular system to employ automatic power control to direct extra power only to those mobiles that are temporarily disadvantaged and to reduce power to those in a favorable situation. In this way, the total downlink power divided by the number of links supported is determined by the mean propagation plus fading loss and not the worst case. The power control algorithm can operate independently on the adjacent channel signals such that they are increasing while the wanted signal power is being decreased. To allow adjacent channel signals to have a 10dB higher power than the wanted signal, it is desirable that the spectral spreading of the modulation be reduced and that adjacent channel energy be reduced from the -18dB to -20dB range to the -28 to -30dB range.

The adjacent channel energy using GMSK results from it being a constant amplitude modulation. Constant amplitude modulation is preferred for transmission from mobile phones as constant-envelope transmitters are simpler and more efficient than non-constant envelope or linear transmitters. There is however no disadvantage in using linear modulation on the satellite downlink, as the active phased-array satellite transponder is in any case adapted to deal with multiple signals, whose composite sum has a varying amplitude. The receivers in GSM mobile phones are furthermore normally adapted to treat the received signal as if it were a linearly modulated signal; this approximation of GMSK to a linearly modulated signal simplifies the receiver design while causing only a small loss of Eb/No performance that is of no consequence in cellular systems. The use of linear modulation for the satellite downlink, for which GSM mobile phone receivers are perfectly adapted, will thus improve the receiver performance compared to transmitting GMSK as well as reducing the adjacent channel energy. The linear modulation that is compatible with GSM receivers is a form of Offset Quadrature Phase Shift Keying (OQPSK). This modulation is generated by applying positive or negative-going signal impulses representing even-numbered data bits to one filter channel (the I-channel) alternately with applying signal impulses representing the odd-numbered data bits to a second filter channel (the Q-channel). The filtered outputs of the I and Q channels then multiplicatively modulate a Cosine and a Sine carrier wave respectively which are then added to form the OQPSK signal. The filter characteristics thus define the transmitted spectrum. GSM uses Gaussian filter shapes defined

by a BT product of 0.3. This restricts the bandwidth and causes some intersymbol interference, which nevertheless is compensated by the equalising receiver. Reducing the BT product to 0.25 for example improves the suppression of adjacent channel energy at the expense of introducing more ISI; this would not be desirable in the GSM system where the equaliser needs to reserve is capability to deal with ISI arising from delayed echos in the land-mobile radio propagation environment caused by reflections from mountains, tall buildings and so-on. The satellite-mobile propagation path suffers less from such delayed echos however, as the path is more nearly line-of-sight; therefore the adjacent channel suppression of the satellite transmissions may also be improved by heavier filtering such as by reducing the BT product of the Gaussian premodulation filters, as well as by using GMSK-compatible linear modulation, both techniques being completely compatible with existing GSM mobile phone reception techniques.

Nevertheless, this only mitigates the effect of adjacent channel powers potentially being higher than the wanted signal due to the operation of dynamic power control, so that the assumptions in the above tables are valid. The result showing that the amount of room for coding in the 32-slot case is marginally insufficient is a potential problem for adopting the 32-slot format as the one and only waveform available.

To achieve the co-channel C/I values required for different amounts of coding, mobiles re-using the same channel must be separated sufficiently on the ground. The aforementioned application which has been incorporated by reference discloses how mobiles may be sorted into groups that satisfy the separation requirement for co-channel operation. Greater C/I's require greater separations, leading to a reduced capacity per unit area for the use of that frequency. Since in the future substantial sums may be paid at auction for the right to use a given amount of frequency spectrum, reducing the capacity served for each frequency bought is not economically desirable. The capacity provided in a given amount of frequency spectrum is however dependent on a combination of the re-use distance needed to achieve the required C/I, and the bandwidth occupied by each signal, and both the C/I requirement and the bandwidth vary contrarily with the amount of coding.

The re-use distance for achieving a given C/I may be shrunk through the use of larger antenna arrays with finer angular resolution, but this increases the cost of the satellite and therefore comparison of different choices needs to be made on the basis of a constant antenna aperture.

The C/I caused by co-channel interference from other antenna beams or directions is a function of the antenna pattern sidelobe characteristics. The sidelobes and thus the interference from adjacent beams may be reduced by tapering the power profile across the array; however, tapered illumination reduces the aperture efficiency and thus the gain compared to uniform illumination. Moreover, the sidelobe level of large phased arrays can be very dependent on phase and amplitude tolerances, which are therefore preferably controlled adaptively as described in the aforementioned patent application. With non-adaptive control, perfect phase and amplitude matching and uniform

aperture illumination, the co-channel C/I is shown in figure 3 as a function of the re-use distance. The distance is given in terms of the -3dB diameter of the main radiation lobe.

The grid spacing for 0.5 and 1dB C/I loss respectively as a function of the number of timeslots (amount of coding) from figure 3 must be greater than the following:

| | 8-slot | 16-slot | 24-slot | 32-slot |
|---|---|---|---|---|
| 0.5dB loss | 1.09D | 1.2-1.9D | 2.05D | 3.3D |
| 1.0dB loss | D | 1.07D | 1.13-1.8D | 1.95D |

CAPACITY (CHANNELS PER 200KHz PER $D^2$)

| | | | | |
|---|---|---|---|---|
| 0.5dB C/I loss | 6.7 | 4.4 - 11 | 5.7 | 2.9 |
| 1.0dB C/I loss | 8 | 14 | 7.4 - 18.8 | 8.4 |

The uncertainty for the 16-slot (0.5dB C/I loss) and 24-slot (1dB C/I loss) cases is due to sandwiching between optimistically counting on a 3dB C/I increase due to DTX and the pessimistic assumption that the peaks in the C/I versus distance separation curves will be eroded due to irregularities in the frequency re-use grid, or by the array radiation diagrams not showing clean nulls in their sidelobe patterns. Use of adaptive array signal processing could tend to produce the higher figure while non-adaptive array processing could tend to produce the lower figure.

A summary of these capacity estimates is plotted in figure 4.

It is clear that a choice of 16 or 24 with a leaning towards the smaller number results in greatest spectral efficiency for a given impact on power efficiency. When account is taken of the extra 0.4dB of coding gain in the use of rate 1/3rd coding (16 timeslot case) as opposed to rate 1/2 coding (24 timelsot case), the power efficiency of the 16-slot case with 1dB degradation due to C/I is equivalent to that of the 24-slot case with 0.5dB degradation. The capacity etsiamtes for these two cases are 14 and 6 for 16 and 24 timeslots respectively however, reinforcing the argument for fewer timeslots and more coding, i.e. the 16-timeslot choice.

Thus, the so-called GSM half-rate TDMA format has been demonstrated to be close to an optimum choice for a satellite communications waveform from both power and bandwidth efficiency viewpoints, although it is used in this invention in a different way from that envisaged in the GSM standards.

The half-rate TDMA format is used in GSM to transmit half the information rate while the format is used in this invention to transmit 1/4 the information rate with twice as much coding. This facilitates the construction of a dual-mode satellite/cellular terminal and opens the possibility to also revert to the GSM half-information-rate speech vocoder in the satellite mode in order to obtain higher quality satellite communications when link margin allows, dropping back to a 4Kilobit vocoder when needed to maintain the link as signal levels become marginal. The option is also open to occasionally transmit only every alternate one of the allocated slots in a nominally 16-slot frame using a coding rate of 2/3rds approximately instead of 1/3rd,

that is to adopt a 32-slot frame, when conditions allow.
A further option is to transmit the allocated slot in a 16-slot frame alternately from a first and a second satellite to obtain path diversity, as described in U.S. patent application no. ............ entitled "Satellite Diversity" (Dent and Ewerbring, filed ..... BDSM case 027545-028) which is also incorporated by reference herein. The bits transmitted by each satellite should preferably be chosen each to constitute rate 2/3rds codes such that either alone could be efficiently decoded, signal quality permitting, while both can be jointly decoded as a rate 1/3rd code when the signal quality of neither alone is adequate. Satellite diversity according to this technique provides improved performance when the signal from each satellite fades independently, as can happen due to the user turning his head and shadowing the signal from one satellite but not the other. As a final option, noting the ability to receive information with a rate 2/3rds code and only using every 32nd slot when signal quality allows, this mode can be useful in certain non-uniform traffic distributions oin order to increase the peak capacity in certain cells when the neighboring cells are lightly loaded. When other cells do not need to use the same frequency channel due to a low traffic demand in those cells, the C/I is improved allowing rate 2/3rds coding and the 32-timeslot format to suffice, thus doubling the capacity in a cell with a high traffic demand. According to one aspect of the invention, the transmission of information every 32th slot using rate 2/3rds coding for example or every 16th slot using twice a much coding may be chosen by the satellite system ata any time and even dynamically without forewarning the mobile phone of the change. An inventive phone receives every 16th slot all the time but determines when that slot contains intended information and if not labels the missing bits as erasures or zero quality value at the input of the error-correction decoder. The system may for example indicate when the information transmitted in a slot is not intended for a particular phone but for another phone by for example using different syncword patterns embedded in the TDMA burst according to figure 2. The different synwords are preferably chosen to be orthogonal patterns to facilitate discrimination.
Even when two conversations to two phones are taking place using odd and even 16-slot frames respectively, i.e. each mobile gets every 32nd slot interleaved with the other mobile, when one speaker is temporarily silent, which happens half the time (DTX), the other mobile may be sent every 16th slot thus providing doubled coding protection for at least half the time to both mobiles.

Figure 5 illustrates TDMA frame structures that can be formed when using 8,16,24 or 32-slot frames. In the 16-slot TDMA frame case, it is also necessary to provide 16 independent SACCH message transmissions. This is done by eliminating the idle frame (13) and forming instead a double-length SACCH frame. The idle frame is no longer necessary, as even if the receiver must scan alternate channels without losing traffic data, the 16-slot format provides one of the original slot periods of free time in every TDMA frame compared to the GSM 8-slot format; thus adequate spare time is available in the format for scanning for other signals if this is desired.

The SACCH frame is also changed to 16 slots in conformity with the speech frames, as this gives a more regular structure than transmitting the additional 8 SACCH messages in the original idle frame positions of the 8-slot structure. The 16-slot frame structure also provides 6, 20mS voice frames or 3, 40mS voice frames of data per 120mS structure repetition period with the same interleaving pattern for all channels.

Unfortunately this desirable feature is difficult to provide with the 24-frame format. When three 8 slot frames are grouped to form a 24-slot TDMA frame, the number of 24-slot frames per 120mS structure period becomes 8. However, 6, 20mS voice frames or 3, 40mS voice frames have to be interleaved over the 8 TDMA frames. Since 8 does not divide by three, the same interleaving pattern cannot be used for all speech frames. On average, a TDMA burst in the 24-slot format must contain 3/4 of a 20mS speech frame, so 1 and 1/3rd bursts are required to accommodate a whole speech frame.

The interleaving pattern can be made regular, i.e. the same for all speech frames and channels, only by increasing the interleaving delay to span a multiple of three TDMA bursts, i.e. 24, which is undesirable because of the attendant increase in speech delay, or by changing the vocoder analysis frame period from 20 or 40mS to 15mS or 30mS, such that a 120mS structure period then contains 8 or 4 vocoder frames which can be distributed between the 8 TDMA frames in a regular manner. Since most available vocoders operate on 20mS or 40mS frames, the 15 or 30mS vocoder frame is not a preferred option.

Figure 5 also illustrates a 32-slot frame structure. This is not necessarily intended to support 32 independent users from a bandwidth efficiency perspective, as that would require 32 independent channels of SACCH messaging. This may in turn require an increase in the structure period to 240mS in order to obtain a 32-slot SACCH frame once in every 52 of the original GSM frame periods. This is not a preferred approach as it introduces a 16mS hiatus in speech transmission that has to be bridged by means of a delay buffer, thus adding extra speech delay. The 32-slot TDMA structure is primarily intended to be regarded as the transmission of a burst allocated in a 16-slot TDMA frame every second frame, when signal quality allows, in order to accommodate more users from a satellite power utilization perspective. The slots not transmitted in one beam can be chosen to coincide with slots that are transmitted in half of the surrounding beams and both adjacent channels in the same beam, thus improving C/I. In otherwords, alternate frequency channels in the same beam transmit in alternate ones of the 16-slot frames, while co-channels in a second beam transmit in slots corresponding to adjacent channels transmissions in the first beam.

If advantage of the 32-slot structure is to be used to improve capacity in heavily loaded cells when traffic distribution is uneven, as described above, the SACCH message can be addressed to one or other phone sharing the same channel in alternate frames by using an odd/even indicator bit in the body of the message.

A rate 1/3rd code can be constructed as a rate 2/6ths code, which generates 6 coded bits for every two information bits that are input. Furthermore, the six coded bits can be distributed by interleaving between two successive TDMA frames such that three of each six occur in both frames and constitute

a good rate 2/3rds code when only alternate frames are used, and which may be regarded as a punctured rate 2/6ths code.
In a punctured rate 2/6ths code, if the 6 coded bits are labelled P1,P2...P6 it need not be so that P1,P2,P3 are assigned to one burst and P4,P5,P6 to the next burst for every information bit-pair, but can for example mean that coded bits P1,P2,P3 for even data bit-pair shifts into the encoder are assigned to the same burst along with P4,P5,P6 for the odd data bit-pairs. This guarantees like performance of the rate 2/3rds code obtained by a mobile receiving either the odd or the even TDMA frames, as all six encoding polyonomials are used in equal amounts in both cases.

A punctured rate 1/3rd code can also be employed, with P1 being assigned to even TDMA frames for even data bits and P2,P3 being assigned to odd frames for even bits, and vice versa. This also guarantees like performance whether only odd frames or even frames are received.

When a 16-slot format is used and alternate frames are transmitted from different satellites, the frames may or may not be transmitted on the same frequency. The GSM format includes sufficient guard times to allow a frequency synthesizer to change frequency between even and odd frames in order to construct a frequency hopping system. Thus satellite diversity can be provided by transmitting the even frames from one satellite on frequencies fo,f2,f4... and in between transmitting the odd frames from another satellite on frequencies f1,f3,f5....

When only one satellite is available, it may transmit only even bursts, only odd bursts, or both, according to the link margin and coding gain needed. When only even bursts are transmitted, the odd bursts may be used for another 16 mobiles and a 32-slot TDMA system is in use. In any duplex conversation, on average one party is silent for half the time, due to the other party talking. Thus even when different mobiles use an even slot and a corresponding odd slot respectively, both slots are available to transmit to each mobile for on average half the time, when the speech for one of the mobiles is temporarily only silence. Moreover, with the use of independent frequency hopping for the odd and even frames it can be arranged that the odd slot corresponding to the even slot allocated to a particular mobile is not always associated with the same other mobile. Thus the probability of the corresponding slot being silent and thus available for enhancing transmission is a random 50% from frame to frame. This is an improvement on the earlier description of using DTX such that each mobile benefits from the other's silence half the time, in that the periods when both are talking and neither benefiting from receiving every frame are not proplongued periods when frequency hopping is employed in this inventive manner.

According to this aspect of the invention therefore, one mobile is allocated a first one out of every 32 slots in a TDMA signal structure on which it will always be given priority to receive information, and a second slot on which it may also receive information if so indicated by an indication contained therein; a second mobile is given priority to receive information on said second slot while being transmitted information also using said first slot when the first mobile does not need priority due to that direction of conversation being temporarily silent. Furthermore, the frequency on which

information is or may be transmitted to a particular mobile is varied every 16 slots by means of a frquency hopping sequence generator, so that the two mobiles paired in the above manner are not always the same two mobiles but are paired with others pseudorandomly from frame to frame.

Using the above aspect of the invention, a mobile receiver cannot predict whether every 16th slot will contain intended information or only every 32nd slot. Consequently the mobile receives both odd and even slots and checks for an identifier to determine if the slot is identified as intended for it or for a different mobile. The GSM format comprises a 26-bit syncword in the middle of the slot which is always the same bit pattern for an intended mobile. The other mobiles that can transmit on that slot may be allocated different syncwords preferably orthogonal to the intended syncword so that their intended information may be easily discriminated.

The GSM standard discloses interleaving blocks of speech data representing 20mS speech segments over eight full-rate frames, using every 8th slot over an 8-frame, 64-slot interval. The speech blocks are diagonally interleaved over this interval with half of a previous block in the first four frames and half of a subsequent block in the second four frames.

In the 16-slot frame format, the same interleaving period comprises only four of the longer frames; two of these are the even frames discussed above and two are odd frames. Each may or may not contain data for the same mobile depending on whether the signal for another mobile is silent or not. Each mobile receives therefore both an odd and an even frame slot and determines if the slot contains data for it. The slots deemed to contain intended data are demodulated to obtain coded bits. The coded bits are in the form of "soft decisions" that comprise quality information related to signal to noise ratio of the bit. Bits not received corresponding to a slot deemed not to contain intended information are given a quality or soft value of zero, corresponding to a symbol erasure. Bits having the erasure indication are said to have been punctured out and the subsequent error correction decoding can save resources by not including punctured or deleted code bits in its decoding process. After deinterleaving, bits originally punctured out in a contiguous block due to the whole slot being non-intended are dispersed between bits of non-zero quality and so the error correction decoder receives many good bits in any section of coded data, thus enabling it to decode the information. In any speech block, two slots will definitely have contained valid data giving three coded bits per two information bits, that is 6 coded bits for four information bits, while half of the other two slots will also contain intended data, giving a further 3 bits on average representing the same four data bits. Thus, the average coding rate obtained is 9 coded bits per four data bits, or better than rate 1/2 coding. The least coding obtained is rate 2/3rds, while the most coding obtained is rate 1/3rd when all four suiccessive slots are directed to the same mobile. The random variation of coding rate between rate 2/3rds, rate 1/2 and rate 1/3rd from one speech frame to another is not of particular significance, as the perceived speech quality is related to the mean speech block error rate, usually called frame erasure rate or FER. Correct decoding of a speech block may be checked by including a Cyclic Redundancy Check code in the block. Blocks detected with the help of the CRC code to have been decoded in error are termed Erased. An

| Uppgjord - Prepared | | Datum - Date | Rev | Dok nr - Doc no |
|---|---|---|---|---|
| Paul W. Dent | 9199907121 | 1994-10-27 | | T/V 94:0020 |
| Godk - Approved | Kontr - Checked | | | File |

erased block, representing a 20mS segment of the speech waveform, is prevented from causing an annoying click or noise burst in the earphone by replacing it with a sound segment previously received correctly. This technique of "bad frame replacement" is disclosed in British Patent no. ........... entitled .............. (Dent, filed ..circa 1980..).
Variations adapted to voice coders that represent speech segments by sets of numerical parameters are also known by the general term "parameter interpolation". Parameter interpolation may be used to bridge over speech frames lost due to errors, and acceptable speech quality for telephone calls is deemed to be obtained when the FER is 5% or less. Thus, providing some frames with more coding and energy than others reduces the FER compared to transmitting only the 32-slot format, and thus allows the quality criterion to be met even though the same number of mobiles has been accommodated in the spectrum as if the 32-format had been used permanently.

The above technique improves principally spectral utilization efficiency by accommodating twice as many users in the same bandwidth while using an increased amount of coding on average per signal, through exploiting discontinuous transmission (DTX). DTX may always be exploited to win 3dB in power efficiency by simply driving the satellite transponder 3dB harder towards saturation to compensate for half the signals being silent.
The active signals get roughly double the power on average by this means.

Attention is now turned to the corresponding uplink format. It is not desirable to increase the number of slots in the uplink TDMA frame when the vocoder bitrate is reduced, as this increases the peak-to-mean ratio of the mobile phone transmitter. Portable battery operated phones are limited in the peak current as well as the mean current that can efficiently be drawn from the battery, owing to the battery's internal resistance, which increases at end-of-life. Therefore the preferred solution for the uplink is to reduce the bandwidth of the transmission or to use more coding, in both cases with the aim of avoiding an increase in peak-to-mean power ratio. As has been seen in the case of the downlink, increasing the bandwidth by use of redundant coding does not necessarily cause a loss in spectral efficiency and capacity, but the reverse, due to a reduction in the frequency re-use distance.

Figure 6 shows the correspondence between up and downlink frequencies and timeslots when using the invention of U.S. patent no. ......... incorporated above (Hybrid access methods), and the uplink comprises 4 timeslots on each of four 50KHz channels in 1:1 association with 16 timeslots on one 200KHz downlink channel. The incorporated invention provides a nominally constant time-duplex spacing between transmit and receive for all channels, which is useful in simplifying the design and operation of the mobile phone.

Just as alternative 32-slot and 16-slot operation modes may be dynamically mixed on the downlink, the invention comprises corresponding alternative 8 and 4-slot modes on the uplink. The motivation and dynamic selection by the mobile to transmit on every 4th uplink slot or every 8th uplink slot is different from the motivation of the system to use 32 or 16 slot formats on the downlink. The system chooses to transmit in the 16-slot format when it is power limited and not bandwidth limited, as the 16-slot format contains

more coding and is thus more power efficient. The satellite's multi-carrier power amplifiers thus need allocate a smaller proportion of their total power to a particular mobile if the 16-slot mode is used.

By contrast, the mobile unit conserves power when transmitting in the 8-slot format rather than the 4-slot format, as it transmits more efficiently using a higher power for half the time than a lower power for twice the time. Since the mobile does not have a multi-carrier power amplifier, its transmitter achieves maximum efficiency at full power. In this mode of operation, the mobile unit is allocated every 4th uplink slot but it may choose sometimes to omit transmission of alternate slots and thereby adopt the 8-slot format, or indeed not to transmit at all if the speaker is temporarily silent (DTX) thus conserving the most power. The mobile also is able to transmit at either full power or any one of a number of progressively lower power levels as further mean to save battery power, as "talk time" between battery charges is of critical interest to the user.

The choice of power level and the use of 4 or 8 slot uplink mode is made by the mobile using a power control algorithm. The preferred power control algorithm comprise both an open loop and a closed loop element, defined by the equation:

EFFECTIVE TRANSMIT POWER LEVEL = CONSTANT - RECEIVED SIGNAL STRENGTH

with the understanding that all quantities are on the logarithmic decibel scale. For example, if maximum effective transmit power level available is 0.5 watts (+27dBm), and the minimum decodable signal strength is -112dBm, the above equation might read:

EFFECTIVE TX POWER (dBm) = (-85) - RECEIVED SIGNAL STRENGTH (dBm)

It can be verified that the above equation sets the effective tranmitter power to the maximum value of +27dBm when the received signal strength is at is minimum useable value of -112dBm. This is based on the reasonable assumption that the uplink path will also be marginal when the downlink path is marginal, thus requiring maximum transmitter power. The closed loop element of the above power control algorithm comprises allowing the fixed network via a land base station or staellite relay station to control the value of CONSTANT used by the mobile to be vlaues other than the exemplary -85 used above. For example if the network controls the power level it allocates to transmit to the mobile terminal, then the signal strength received at the mobile varies with the allocated downlink power level even if the downlink propagation path has a constant attenuation. The fixed network should therefore from time to time by means of the SACCH message facility for example command the mobile to use different values of CONSTANT in dependence on the mean level of downlink power tranamitted for that mobile. Alternatively or additionally to the relatively slow FACCH mechanism for performing closed loop power control, a faster feedback method can be employed which allocates one bit of transmitted data to signify to the mobile that it shall step up or step down its transmitter power by a given amount. Considering propagation delays in a satellite system however, the speed advantage over the use of SACCH may not be great.

An alternative power control system may be used which provides the mobile unit with more control over the power used on both uplink and downlink. Usually a system operator does not wish to place control in the hands of mobile units, however, in the case of satellite communications where billing rates can be adjusted according to the satellite power consumed, the problem of potential abuse is less worrysome. It is in any case normal practice for the mobile unit to report, using the SACCH message facility, the signal quality or strength it recently received on the downlink. The network station receives the information relayed over the satellite from all active mobiles and then reallocates downlink power to attempt to equalise their received signal qualities based on the reported signal quality. Thus already in existing are it can be said that the mobile is at least partly in control of the signal strength it receives.

By making the algorithm that the network uses to allocate downlink power based on reported signal quality a deterministic or predictable one, the mobile can predict ahead of time after sending a signal strength report what power the network will allocate on the downlink to that mobile in some future frame delayed by the loop propagation delay to and from the satellite and ground network. Thus the mobile can itself adjust the value of CONSTANT to compensate for future changes in the downlink power.

Irrespective of which of the above variations of power control algorithm is used, the required transmit power is first determined as a numerical value inside the control processor of the mobile terminal. The power level computed is then used to command the duty factor (4 or 8 slot mode) and the burst power level of the mobile transmitter. If the maximum power level is demanded, the mobile uses the 4 slot mode at maximum burst power if the network has previously indicated that the mobile may transmit on every 4th slot; otherwise the 8-slot mode is used at maximum burst power.

For power levels between maximum and 3dB below maximum, the 4-slot mode is also used with a power level redcued by up to 3dB. For power levels of 3dB or more lower than maximum, the 8-slot is used either at full burst power (corresponding to a demand for full power - 3dB) or at less than full power. As an alternative the mobile can alternate between transmitting the 4-slot and the 8-slot mode, effectively deciding on a frame by frame basis in depedence on the received downlink signal quality in the previous downlink burst whether it shall transmit also on a second slot out of every 8 or only on one of the 8.

Thus it can be seen that the preferred power control method is to employ duty factor variation between 1/4 and 1/8 to effect the top 3dB of power control range, as this may be accomplished while leaving the transmitter in its most efficient full-power regime during the transmitted bursts. When more than 3dB reduction of power is called for, the preferred approach is to have the mobile transmit at the highest and most efficient burst power level but with the lower or lowest duty factor available. In a satellite or cellular system, considerable link margin must be available to cope with fading and shadowing that can occur depending on the mobile's location or movements, but maximum

margin is not required all the time. Dynamic power control as disclosed above allows the mobile unit to achieve considerable battery power savings on average by only using high power or duty factor when needed.

The use of 1/8 alternative 1/4 duty factor as an uplink power control means is different from its use to double capacity in bandwidth-limited cells. Both modes may however be used in the same system as long as the system indicates to a mobile which mode it shall assume from time to time, such as at call set up or later during a call by using SACCH or FACCH messages or other mechanism. For example, the mobiles in a heavily loaded cell can be divided by the network into a group that is signal-strength-disadvantaged, for whatever reason, that shall be allowed to use 4 or 8-slot mode dynamically, and a group that is more favourably disposed that can get by with 8-slot operation, thus doubling the capacity for that group. Regrouping can take place dynamically should a mobile unit change from being favourably located to being unfavourably located. When two satellites illuminate the same area, both can attmept to receive each mobile transmission and thus improve uplink signal quality by satellite diversity as disclosed in the aforementioned application taht has been incorporated by reference. Thus only mobiles that are not favourably disposed to be received by either satellite need to belong to the group that transmits every 4 timeslots.

It may also be noted that two-way conversation generally comprise a flow of speech traffic in only one direction at a time, thus the benefits of of 16 versus 32-slot downlink usage should not be required by the same mobile at the same time as the benefits of 4 versus 8 slot uplink usage. Consequently, if a first mobile detects that it is receiving information on the downlink in every 16th slot, it indicates that a second mobile with which it is instantaneously paired is silent on the downlink and therefore probably active on the uplink. The first mobile should then only adopt the 8-slot uplink mode or be silent (DTXed). On the other hand, detecting in the first mobile that only one out of 32 timeslots was transmitted to it indicates that the downlink of the second mobile was also active and that transmission on every 1 out of 8 or 1 out of 4 uplink slots may be permitted. In thus way, uplink clashes are confined to those times when the downlink is the principal direction of speech activity.

The above inventive use of dynamic TDMA slot allocation combined with demodulation and decoding that can automatically detect whether received information was intended or not and discard it or use it accordingly has many advantages as have been described in the above specification but which are meant to be exemplary and not limiting of the scope of the invention which is defined by the following claims.

## 5. CLAIMS

I claim:

1. A dual-mode mobile communications unit adapted in one mode for receiving from and transmitting to a land-based network station Time Division Multiple Access signal bursts in an allocated timeslot of a TDMA frame having a first plurality of timeslots for transmission and reception and adapted in another mode for transmitting to an orbiting satellite or airborne relay station Time Division Multiple Access signal bursts in an allocated timeslot of a TDMA frame having a second plurality of timeslots for transmission and said second plurality is lower than said first plurality.

2. A method of communicating information using Time Division Multiple Access and adaptive transmission and reception comprising:-

   -Using TDMA burst transmission means for transmitting signal bursts to a TDMA receiving means wherein said transmission means codes said information and transmits coded information to said receiving means using one or alternatively two timeslots of a plurality of timeslots in a repetitive TDMA frame period;

   -Using TDMA receiving means to receive both of said two timeslots whether or not said transmitting means has transmitted using said one or said two timeslots and classifying received signals as intended or non-intended;

   -assembling successively received signals classified as intended into a block for decoding;

   -decoding said block to reproduce said information.

3. A method of communicating information according to claim 2 wherein said transmission means codes said information using a convolutional code.

4. A method of communicating information according to claim 2 wherein said transmission means inserts a known symbol pattern into said coded information for transmission.

5. A method of communicating information according to claim 4 wherein said receiving means classifies said received signals as intended or not intended based upon detection of said known pattern.

6. A method according to claim 2 wherein said burst transmission means uses said alternative of two time slots when determined to be required by said TDMA receiving means in order to obtain

| Uppgjord – Prepared | | Datum – Date | Rev | Dok nr – Doc no |
|---|---|---|---|---|
| Paul W. Dent | 9199907121 | 1994-10-27 | | T/V 94:0020 |
| Godk – Approved | Kontr – Checked | | | File |
| Paul W. Dent | | | | |

good decoded information quality and said burst transmission means uses one time slot when that alone is determined to provide good decoded information quality.

7. A method according to claim 2 wherein said burst transmitter means when using said alternative of two timeslots repeats in a second of said two timeslots transmissions of coded symbols already transmitted in a first of said two timeslots.

8. A method according to claim 2 wherein said burst transmitter means when using said alternative of two timeslots transmits in the second of said two timeslots at least some information symbols coded in a second manner that were already transmitted in the first of said two timeslots coded in a first manner.

9. A method according to claim 8 in which said first manner of coding comprises selecting half of the output bits of an error correction coder when the input bits to said coder are said at least some information symbols and said second manner of coding comprises selecting the other half of said output bits.

10. A method according to claim 2 in which said TDMA burst transmission means when using said alternative of two-timeslot transmission comprises a first TDMA transmission means for transmitting on the first of said two timeslots and a second, separate TDMA transmission means for transmitting on the second of said two timeslots.

11. A method according to claim 10 in which said first and second TDMA transmission means are not colocated.

12. A method according to claim 10 in which said first and second TDMA transmission means respectively comprise a first and a second orbiting satellite relay station.

13. A method according to claim 10 in which said first and second TDMA transmission means respectively comprise a first and a second airborne relay station.

14. A method according to claim 10 in which said first and second TDMA transmission means respectively comprise a first and a second cellular base station transmitter.

15. A method according to claim 14 in which said first and second cellular base station transmitter are connected to separate sector inputs of a sectorized transmit antenna.

16. A method according to claim 11 in which said first and second TDMA transmission means comprise different cellular base stations and sites.

| Uppgjord – Prepared | | Datum – Date | Rev | Dok nr – Doc no |
|---|---|---|---|---|
| Paul W. Dent | 9199907121 | 1994-10-27 | | T/V 94:0020 |
| Godk – Approved: Paul W. Dent | Kontr – Checked | | | File |

17. A method according to claim 10 in which said alternative two-slot transmission is employed when said TDMA burst receiving means is located in a handover regime between said first and second TDMA burst transmission means.

18. TDMA communication apparatus with improved transmitter power level control comprising:-

-Burst receiving means for receiving TDMA signal bursts in an allocated receive timeslot of a repetitive TDMA frame period and measuring received signal strength;

-Burst transmission means for transmitting TDMA signal bursts in an allocated transmit timeslot of a repetitive TDMA frame period under control of a power control means;

-transmit power control means for computing a desired effective burst transmission power level to compensate for propagation path changes based on said measured received signal strength and controlling said burst transmission means to transmit a signal burst at a controlled power level in said allocated transmit timeslot in each of said TDMA frame periods or alternatively to skip transmission in certain frame periods such that the combination of said controlled power level with the average fraction of frames transmitted provides an effective communication transmit power level equal to said computed desired power level.

19. TDMA communications apparatus according to claim 18 in which said certain frame periods are the odd-numbered frame periods.

20. TDMA communications apparatus according to claim 18 in which said certain frame periods are the even-numbered frame periods.

21. TDMA communications apparatus providing an improved transmit and receive diversity operation comprising:-

-Coding means for coding an information bitstream such that for every information bit input to said coder a plurality of coded output bits are generated;

-first TDMA burst transmission means for transmitting half of said plurality of coded bits using a first TDMA timeslot of a repetitive frame period and a first sequence of radio channel frequencies;

-second TDMA burst transmission means for conditionally transmitting the other half of said plurality of coded bits using a second timeslot of a repetitive frame period and a second sequence of radio channel frequencies;

-TDMA receiving means for receiving on said first timeslot on a frequency in said first frequency series alternately with receiving on said second timeslot on a frequency in said second series and producing demodulated output signals to a decoder;

-decoding means for classifying said output signals received respectively in said first and said second timeslots as intended or non-intended and decoding signals classified as intended to reproduce said information bitstream.

22. The TDMA communications apparatus according to claim 21 in which said conditional transmission by said second TDMA burst transmission means is based on the signal quality decoded by said TDMA receiving means.

23. A portable satellite communications terminal using time division multiple access comprising:-

-TDMA burst receiving means for receiving and analysing signal bursts received in a receive timeslot of a repetitive TDMA frame period and producing decoded information and a signal quality indication;

-TDMA burst transmitting means for coding information and said signal quality indication to produce coded signal bursts for transmission and transmitting coded bursts in an allocated transmit timeslot of even-numbered TDMA frames and conditionally transmitting coded bursts in an allocated timeslot of odd-numbered TDMA frames in dependence on said signal quality indication.

24. A portable satellite communications terminal using time division multiple access comprising:-

-TDMA burst receiving means for receiving and analysing signal bursts received in a receive timeslot of a repetitive TDMA frame period and producing decoded information containing mode commands;

-TDMA burst transmitting means for coding information and said signal quality indication to produce coded signal bursts for transmission and transmitting coded bursts in an allocated transmit timeslot of even-numbered TDMA frames and conditionally transmitting coded bursts also in an allocated timeslot of odd-numbered TDMA frames in dependence on said mode commands.

25. A portable satellite terminal according to claim 24 further equipped to change the radio channel frequency of said TDMA burst receiving means between successive TDMA frames.

| Uppgjord – Prepared | | Datum – Date | Rev | Dok nr – Doc no |
|---|---|---|---|---|
| Paul W. Dent | 9199907121 | 1994-10-27 | | T/V 94:0020 |
| Godk – Approved | Kontr – Checked | | | File |

26. A portable satellite terminal according to claim 24 further equipped to change the radio channel frequency of said TDMA burst transmission means between said even and odd-numbered frames.

27. A portable satellite terminal according to claim 24 further equipped to maintain timing synchronisation of said TDMA burst receiving means independently for receiving signal bursts in respectively even and odd-numbered frames.

28. A portable satellite communications terminal using time division multiple access comprising:-

-TDMA burst receiving means synchronised by synchroniusing means to receive and analyse signal bursts received in allocated receive timeslots of a repetitive TDMA frame period and to produce decoded information and a signal quality indication;

-synchronising means for synchronising said TDMA burst receiving means independently for even-numbered and odd-numbered ones of said TDMA frame periods;

-TDMA burst transmitting means for coding information and said signal quality indication to produce coded signal bursts for transmission and transmitting coded bursts in allocated transmit timeslots in said repetitive TDMA frame period.

29. A communications system for providing a communications service to a plurality of first stations with the aid of a network of relay stations each of said relay stations comprising:-

-coding means for coding data for transmission to said first stations to produce signal bursts;

-TDMA burst transmission means for relaying said signal bursts to each of said first stations using respective timeslots of a repetitive TDMA frame period such that each of said first stations receives signal bursts in its respective timeslot;

-detection means to detect when temporarily no data is required to be sent to one of said first stations;

-means controlled by said detection means to transmit signal bursts intended for another of said first stations in the timeslot designated for said one of said first stations temporarily receiving no data as well as transmitting signal bursts in said another station's designated timeslot.

30. A communications system for relaying digitally coded voice signals to each of a plurality of remote stations comprising:-

-voice coding means for digitizing and coding voice signals to produce frames of digital data containing a fixed number of bits representing a segment of the voice signal over a time period and producing an associated voice/no voice flag;

-error correction coding means for coding each of said voice frames to produce a first coded symbol block and a second coded symbol block each representing a respective one of said speech frames;

-TDMA burst transmission means for relaying said first coded symbol blocks to respectively intended remote stations using a respectively allocated timeslot in a repettive TDMA frame period;

-control means for controlling said burst transmission means to replace a first coded symbol block when said associated flag indicates the no-voice condition by a second coded symbol block intended for a different remote station;

31. The system according to claim 30 further comprising:-

-remote station TDMA burst receiving and decoding means for receiving said coded symbol blocks in a respectively allocated timeslot and an alternate slot and detecting whether said symbol block received in said allocated slot is an intended one of said first coded symbol blocks.

32. The system according to claim 31 in which said remote station TDMA burst receiving and decoding means further detects detecting whether said symbol block received in said alternate slot is an intended one of said second coded symbol blocks.

33. The system according to claim 32 in which said TDMA burst decoding means upon detecting an intended first coded symbol block and an intended second coded symbol block jointly decodes both blocks to give an enhanced probability of correctly reproducing said voice signal segment else decodes only said first coded block if detected to be intended and said second coded block is not detected to be intended.

34. An improved method of radio communications between a network comprising at least two relay stations and a plurality of remote stations using frequency hopping time-division multiple access including:-

-transmitting signals from one of said relay stations to a specific one of said remote stations in a designated timeslot of a repetitive TDMA frame period.using a radio channel selected for each successive frame period using a first frequency hopping sequence generator;

-transmitting signals from another of said relay stations to said specific one of said remote stations in a designated alternate timeslot of said repetitive TDMA frame period using a radio channel selected for each successive frame period using a second frequency hopping sequence generator.

35. A method of comnmunication according to claim 34 in which said first and second frequency hopping sequence generators select frequencies from different frequency sets.

36. A method of communications according to claim 34 in which said first and second frequency hopping sequence generators select frequencies orthogonally from the same set.

37. A method of communications according to claim 34 in which said first frequency hopping sequence generator generates a number of mutually orthoginal sequences and said successively selected radio channels for said specific remote station belong to one of said orthogonal sequences and the other orthogonal sequences are used by other remote stations.

38. A time division multiple access format for transmitting traffic and signalling data between a network station and a plurality of remote stations comprising:-

-dividing a superframe period into an odd number of TDMA frame periods;

-dividing each TDMA frame period into an even number of timeslots;

-using said even number of timeslots in one of said odd number of TDMA frames for transmitting signalling information addressed respectively to a corresponding even number of remote stations and the remaining TDMA frames in said superframe for transmitting traffic information;

-dividing said remaining number of frames used for transmitting traffic information into a first group of traffic frames and a second group of traffic frames;

-using said even number of timeslots in said first group of traffic frames to transmit traffic data to a corresponding even number of remote stations and using the timeslots in said second group of traffic frames to transmit data to a corresponding number of other remote stations.

39. A method according to claim 38 in which said signalling information comprises an indication whether said information is addressed to one of said remote stations for which traffic information was transmitted using a timeslot from each of said first group of traffic frames or for one of said remote stations for which traffic information was sent using a timeslot in said second group of traffic frames.

40. An adaptive time division multiple access format for transmitting traffic and signalling data between a network station and a plurality of remote stations comprising:-

-dividing a superframe period into an odd number of TDMA frame periods;

-dividing each TDMA frame period into an even number of timeslots;

-using said even number of timeslots in one of said odd number of TDMA frames for transmitting signalling information to said remote stations and the remaining TDMA frames in said superframe for transmitting traffic information;

-dividing said remaining number of frames used for transmitting traffic information into a first group of traffic frames and a second group of traffic frames;

-using one of said even number of timeslots in said first group of traffic frames to transmit traffic data to a first of said remote stations and using the corresponding timeslot in said second group of traffic frames to transmit data to another remote station alternatively in dependence on communications signal quality using both said timeslot in said first group of frames and said corresponding timeslot in said second group of frames for communicating with the same remote station.

41. A dual-mode satellite/cellular radio telephone system using TDMA comprising mobile stations, cellular network stations and satellite relay stations for transmitting TDMA signal bursts including the method of:-

-transmitting bursts in said TDMA format intended for a specific mobile station at a first repetition rate from a cellular network station alternatively at a second repetition rate being a sub-multiple of said first repetition rate from a satellite relay station;

-dividing bursts transmitted at said first repetition rate into superframe groups each comprising traffic bursts, a Slow Associated Control Channel burst and an idle frame not containing a transmitted information burst;

-using said idle frame at said specific mobile station to change its TDMA burst receiver channel frequency to that of a neighboring cellular network station or to the channel frequency of a satellite relay station and analysing the signal received on that channel frequency.

42. A communications system according to claim 41 in which said method further comprises:-

-reporting from said mobile station to said network station the results of said signal analysis.

43. A communications system according to claim 41 in which said method further includes communicating said neighboring channel frequencies and said satellite relay station channel frequency from a network station to said mobile station.

44. The method according to claim 42 in which said reported signal analyses are used at said network station to determine when said mobile station shall be switched to receive data from a satellite relay station instead of one of said network stations.

more claims.....

45. An improved method of capacity allocation in a TDMA satellite or cellular communications system comprising a number of areas containing mobile subscriber terminals each area being served by a respective antenna beam, said method including:-

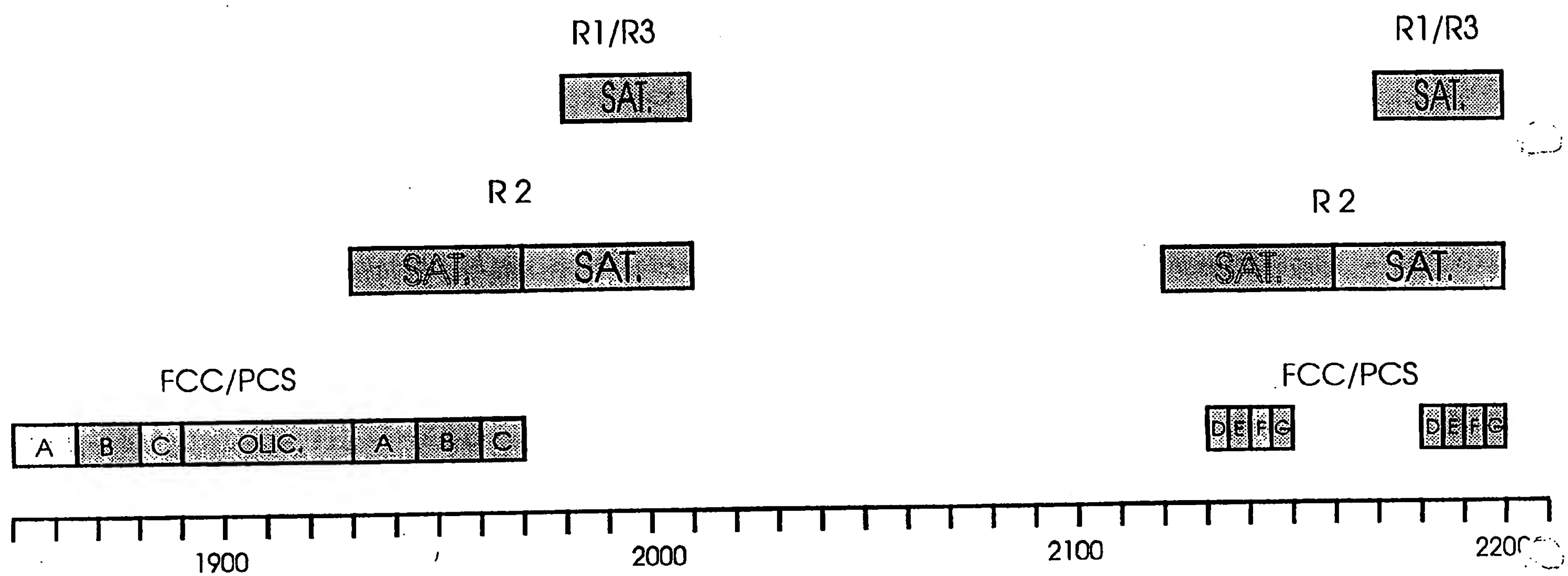
-allocating each mobile terminal in each area a timeslot in a each TDMA frame period and a channel frequency in a frequency set for receiving data whereby mobiles in the same area receive different timeslot and frequency allocations and mobiles in different areas may share the same timeslot and frequency allocation;

-transmitting coded information bits to a mobile terminal in a given area using a respectively allocated timeslot in every one of said TDMA frame periods when the total number of mobiles so served in said given area is similar to the number of mobiles served in surrounding areas and close to a mean number of mobiles expected in each area;

-transmitting coded information bits to a mobile terminal in a given area using a respectively allocated timeslot in every alternate one of said TDMA frame periods and using the corresponding timeslot in the other half of the frame periods for transmitting coded information to a different mobile terminal when the number of mobiles so served in said given area is substantially greater than said mean value and the number of mobiles served in adjacent areas is on average less than said mean value.

FIGURE 1

ALLOKERING AV SATELLIT- RESP. PCS-FREKVENSER

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8 OCT 94

FIGURE 5: 8, 16, 24 AND 32-SLOT FORMATS

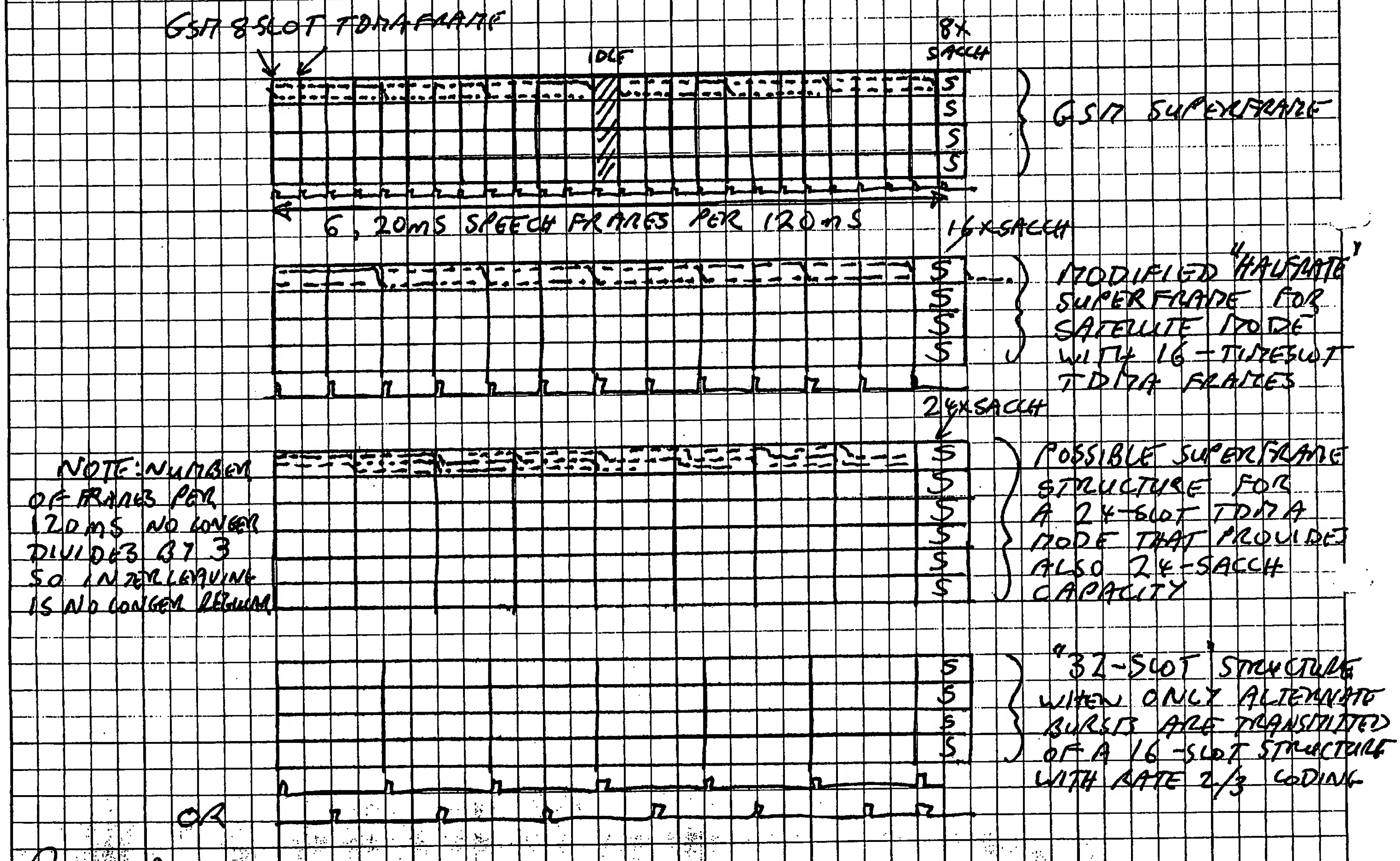


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FIGURE 3

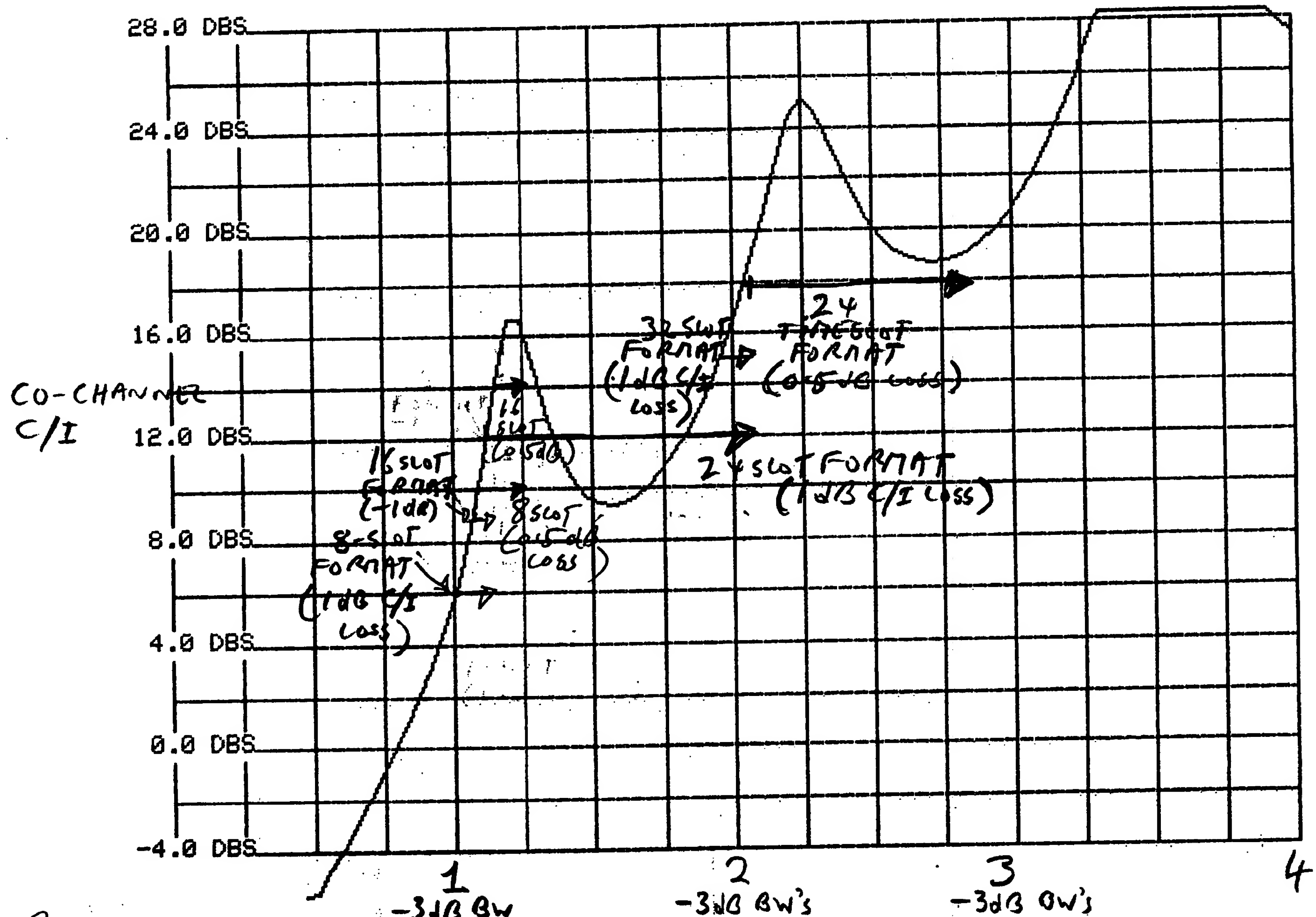


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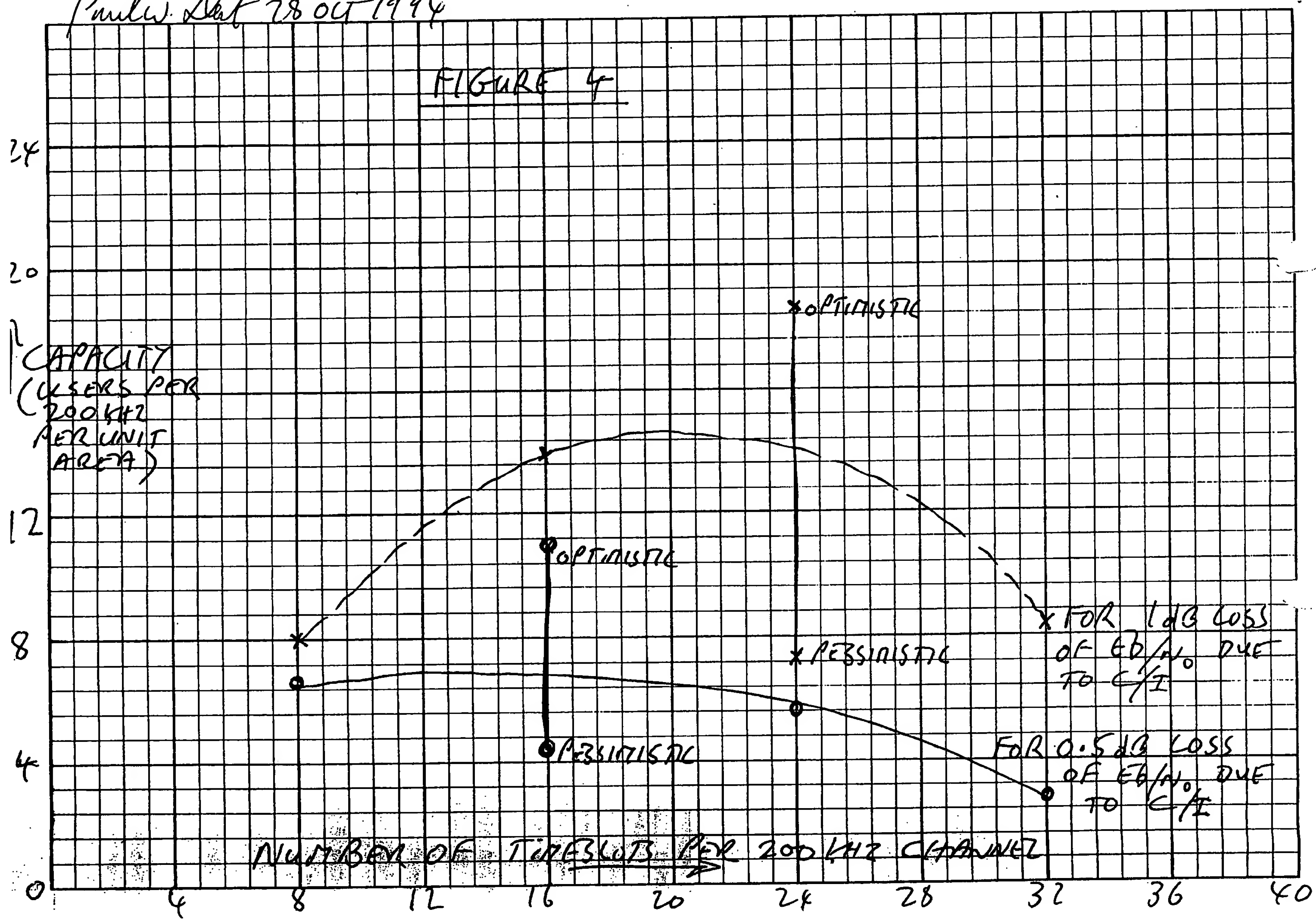
FIGURE 4



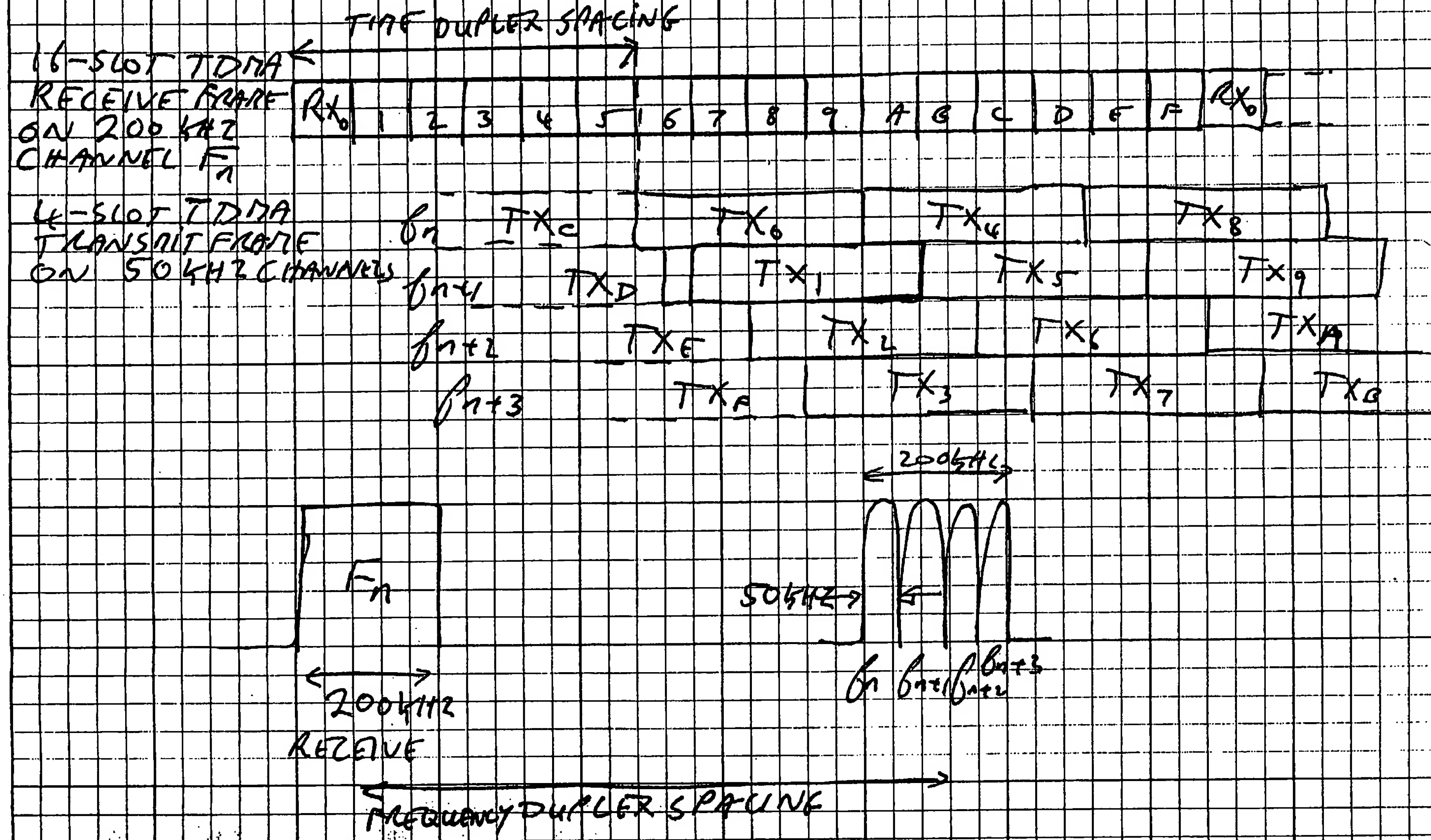



FIGURE 6: CORRESPONDENCE BETWEEN UP AND DOWNLINKS

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